Cambium C3VoIP-150 Installation Manual

Version 1.0

Revision Record

Version	Date	Description	Applicability
Version 1.0	May 2014	First edition	Version 1.0 for Cambium Networks

Contents

Contents 3
1. Introduction 1
    1.1 Package Contents 1
2. Product Overview 2
    2.1 Hardware Specification 2
    2.2 Front View and LEDs Introduction 2
    2.3 Back View and Interface Introduction 3
3. Installation 4
    3.1 Connection topography 4
    3.2 Installation Steps 4
4. IVR 5
    4.1 Ways to Configure 5
    4.2 Start IVR 5
    4.3 IVR Description 5
    4.4 Notice 8
5. Parameter Introduction 9
    5.1 Password 9
    5.2 URL format 9
6. Login to the WEB Server 10
    6.1 Login WEB via the LAN port 10
    6.2 Login WEB via the WAN port 10
    6.3 WEB Interface Introduction 11
7. Configuration from the WEB page 12
    7.1 Status 12
        7.1.1 Basic 12
        7.1.2 Syslog 13
    7.2 Network 14
        7.2.1 RADIO 14
        7.2.2 LAN 16
        7.2.3 MAC Address Clone 17
        7.2.4 VPN Settings 17
        7.2.5 DMZ 18
        7.2.6 DDNS 18
        7.2.7 Port Setting 19
        7.2.8 Multi WAN 19
    7.3 SIP 20
        7.3.1 SIP Settings 20
        7.3.2 VoIP QoS 20
    7.4 FXS1 21
        7.4.1 SIP Account 21

7.4.2 Preferences 24
7.4.3 Dial Plan 26
7.4.4 Black List 27
7.4.5 Call Log 28
7.5 FXS2 29
7.6 Administration 30
7.6.1 Management 30
7.6.2 Security 32
7.6.3 Firmware Upgrade 32
7.6.4 Provisioning 33
7.6.5 SNMP 35
7.6.6 TR069 35
8. Functions 37
8.1 Making Calls 37
8.2 Call Waiting 37
8.3 Call Hold 37
8.4 Call Transferring 37
8.4.1 Blind Transfer 37
8.4.2 Attended Transfer 37
8.5 3-way conference call 38
8.6 Call Forwarding 38
8.7 Direct IP calls 38
8.8 Speed dialing 38
8.9 Hotline 38
8.10 Daylight Saving Time 39
8.11 Upgrade Firmware 39
8.12 Password Control 40
8.13 Web Access 40
8.14 System log 40
9. Software Features 41
10. Third Party Software 42
10.1 OpenSSL 42

1. Introduction

Thank you for choosing the C3VOIP-150 VoIP Gateway. The C3VoIP-150 adapter allows you to make IP Phone calls using your broadband connection.

This manual provides basic information on how to install and connect C3VOIP-150 VoIP Gateway to the Internet. It also includes features and functions of C3VOIP-150 VoIP Gateway components and how to use it correctly.

Before you can connect the C3VoIP-150 to the internet and use it, you must first have a high-speed broadband connection with a Cambium PMP product, such as PMP450 or ePMP AP that is always connected to Cambium PMP SM.

The C3VOIP-150 VoIP Gateway is a stand-alone device, which requires no PC to make Internet calls. This C3VoIP-150 guarantees clear and reliable voice quality on the internet, which is fully compatible with the SIP industry standard and is able to interoperate with many other SIP devices and software on the market.

1.1 Package Contents

User package includes:

- One C3VOIP-150 VoIP ATA Gateway
- One power adapter
- One Ethernet cable

If any of the above items are damaged or missing, please contact your reseller for replacement.

2. Product Overview

2.1 Hardware Specification

Item	Specification
Power Adapter	Input: 100~240V AC, 50~60 Hz Output: DC 12V, 2A
CPU	RTL8972@240 MHZ
Ports	Radio port (WAN Port) and LAN port
Operating Temperature	5～45℃(41～113°F)
Storage Temperature	-25～85℃(-13～185°F)
Relative Humidity	10～90% (No condensing)
Dimension (L×W×H)	152×119×28mm
Weight (packaging included)	366g
Certification	CE / FCC / RoHS

2.2 Front View and LEDs Introduction

Figure 1 Front View

Table 1 LED description

LED indicator	Colour	Status	Description
Phone 1/2	Green	Solid	Registered and no call
		Blinking at 1 Hz	In hook-off or ringing status
		Solid	Linked
LAN	Green	Blinking at 10 Hz	Data transmission
		Light Off	Unlinked
Radio (WAN)	Green	Solid	Linked
		Blinking at 10 Hz	Data transmission
		Light Off	Unlinked
Power	Green	Blinking at 1 Hz	The last provision has failed, re-trying
		Blinking at 10 Hz	The device is upgrading

2.3 Back View and Interface Introduction

Figure 2 Back View

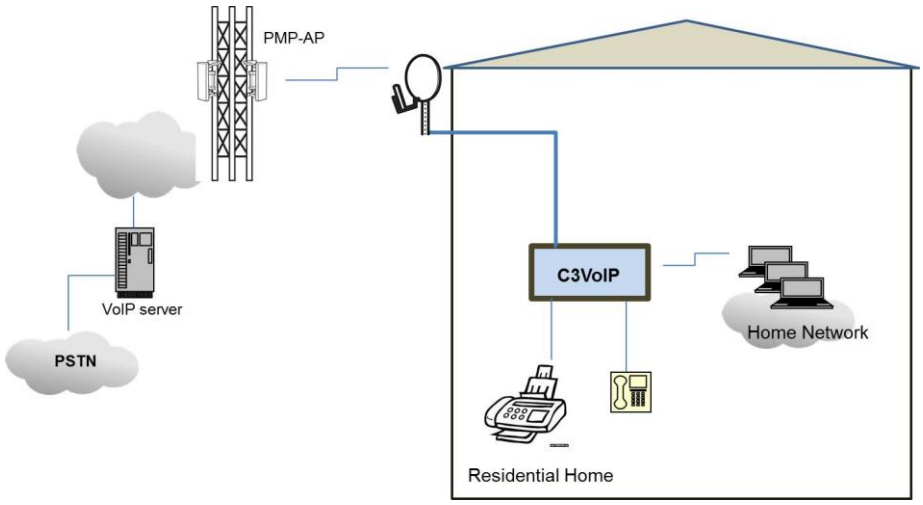
Table 2 Port description

Port	Name	Description
	PHONE1 (RJ11)	Connect to traditional phone or FAX Machine
	PHONE2 (RJ11)	Connect to traditional phone or FAX Machine
	RST	Press 10s to set C3VOIP-150 factory settings.
	LAN (RJ-45)	Connect to PC or build a small LAN or Wi-Fi network
	Radio (WAN) (RJ-45)	Connect to Cambium PMP or ePMP Remote module and provide Cambium proprietary PoE to the radio The Radio port provides WAN connection. The WAN port and Radio port in this document mean the same port of C3VoIP-150.
	Power Port	DC(12V/2A)

3. Installation

This chapter explains the installation procedure of C3VOIP-150.

3.1 Connection topography



3.2 Installation Steps

1. Insert one end of the Ethernet cable into the Radio port at the back of the C3VOIP-150 and the other end into the cable that leads to the PMP remote radio module

> **Note:**
>
> Please refer to one of the SM installation guide(s) for information on how to install your SM.

2. Connect the LAN port at the back of C3VOIP-150 to your conventional telephone, using a standard phone cable
3. If there is a need to setup a small LAN network at home, the C3VOIP-150 must work in router or bridge mode so that users can access to the Internet through C3VOIP-150. Then you must connect your PC or LAN connection equipment (e.g. Ethernet switch or Wi-Fi router) to the LAN port at the back of the C3VOIP-150 using an Ethernet cable. (Step 3 is optional and depends on your needs)
4. Connect the power adapter to the power port at the back of C3VOIP-150 and then plug the power adapter into a wall outlet or power strip. The Power LED of C3VOIP-150 glows to indicate the correct operation. Once the SM is powered on, it may take some time for it to register onto the AP and provide the INTERNET service.

Warning:

Do not attempt to use different power adapters or cut off power supplies during configuration or updating C3VOIP-150 VoIP ATA Gateway. Using other different adapter can damage C3VOIP-150 VoIP Gateway and void the manufacturer warranty.

Warning:

If there is a power outage, E-911 WILL NOT Work.

4. IVR

4.1 Ways to Configure

C3VOIP-150 supports three modes of configuration that can be used:

- Use the IVR method
- Use the web browser method（recommended）
- Use the provisioning method

4.2 Start IVR

Customer can use the IVR function by referring to the following steps:

1. Connect an analog phone to the C3VOIP-150's phone port
2. On the phone dial in ******** to start IVR, then the C3VOIP-150 prompts :
   "Please enter your option, one WAN Port ……".
3. Choose the operation code from 0 to 9, then C3VOIP-150 reports the contents. For details, see Table 3.
4. After every successfully operation, C3VOIP-150 displays the message **"Please enter your option, one WAN Port ……"**.

4.3 IVR Description

Below is the command description list:

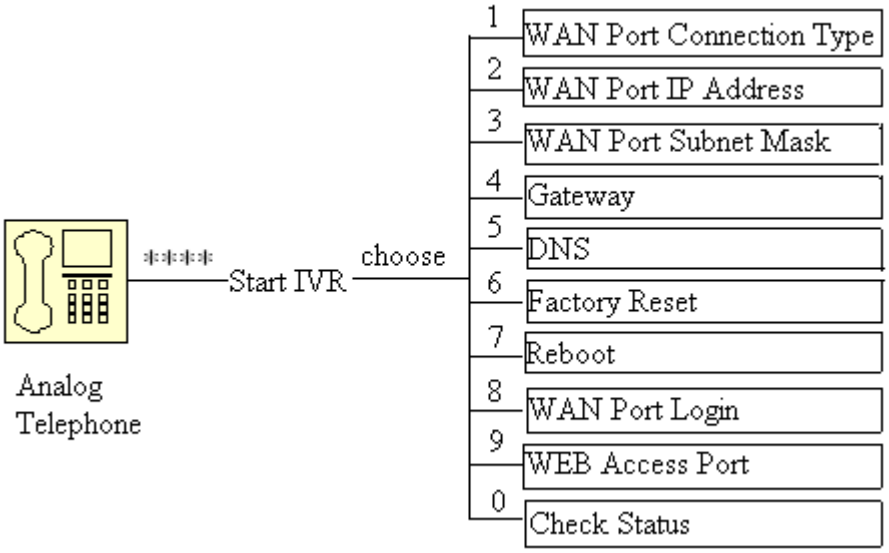


Table 3 Code description:

Operation code	Contents
1	1. On the phone, dial in ******** to start IVR 2. Choose "1". C3VOIP-150 reports the current WAN port connection type 3. When prompted "Please enter password", user must input the password with end char # if user wants the configuration to be a WAN port connection type. ✧ The password in IVR is same as the one of WEB logins. User can use phone keypad to enter password directly, and the matching table is in Note 4. ✧ For example: WEB login password is "admin", so password in IVR is "admin" too. So the user dials "23646" to access and then configure RADIO connection port. If the password is correct, "Operation Successful" is displayed. 4. Choose the new WAN port connection type from 1.DHCP and 2.Static 5. If changes are successful, "Operation Successful" message is displayed. The C3VOIP-150 pops the message **"Please enter your option, one WAN Port ……"** ✧ Note: Add "#" to assume after input password and selected new WAN port connection type ✧ To quit at any point, press "*"
2	1. On the phone, dial in ******** to start IVR 2. Choose "2", and C3VOIP-150 report current WAN Port IP Address 3. Input the new WAN port IP address and with the end char #, ✧ using "*" to replace ".", user can input 192*168*20*168 to set the new IP address 192.168.20.168 ✧ press # key to indicate that you have finished 4. Report "operation successful" if user operation properly. ✧ Note: If you want to quit by the wayside, press "**".
3	Step 1. On the phone, dial in ******** to start IVR Step 2.Choose "3", and C3VOIP-150 report current WAN port subnet mask Step 3.Input a new WAN port subnet mask and with the end char # ✧ using "*" to replace ".", user can input 255*255*255*0 to set the new WAN port subnet mask 255.255.255.0 ✧ press # key to indicate that you have finished 3) Report "operation successful" if user operation properly. ✧ Note: If you want to quit by the wayside, press "**".
4	Step 1. On the phone, dial in ******** to start IVR Step 2.Choose "4", and C3VOIP-150 report current gateway Step 3.Input the new gateway and with the end char # ✧ using "*" to replace ".", user can input 192*168*20*1 to set the new gateway 192.168.20.1 ✧ press # (pound) key to indicate that you have finished 3) Report "operation successful" if user operation properly. ✧ Note: If you want to quit by the wayside, press "**".

5	Step 1. On the phone, dial in ******** to start IVR Step 2.Choose “5”, and C3VOIP-150 report current DNS Step 3.Input the new DNS and with the end char # ✧ using “*” to replace “.”, user can input 192*168*20*1 to set the new gateway 192.168.20.1 ✧ press # (pound) key to indicate that you have finished 3) Report “operation successful” if user operation properly. ✧ If you want to quit by the wayside, press “**”.
6	Step 1. On the phone, dial in ******** to start IVR Step 2.Choose “6”, and C3VOIP-150 report “Factory Reset” Step 3.Prompt "Please enter password", the method of inputting password is the same as operation 1. ✧ If you want to quit by the wayside, press “*”. Step 4.Prompt “Operation successful” if password is right and then C3VOIP-150 is set to factory setting. Step 5.Press “7” reboot to make changes effective.
7	Step 1. On the phone, dial in ******** to start IVR Step 2.Choose “7”, and C3VOIP-150 report “Reboot” Step 3.Prompt "Please enter password", the method of inputting password is same as operation 1. Step 4.C3VOIP-150 reboots if password is right and operation is properly.
8	Step 1. On the phone, dial in ******** to start IVR Step 2.Choose “8”, and C3VOIP-150 report “WAN Port Login” Step 3.Prompt "Please enter password", the method of inputting password is same as operation 1. ✧ If you want to quit by the wayside, press “*”. Step 4.Report “operation successful” if user operation properly. Step 5.Prompt “1enable 2disable”,choose 1 or 2, and with confirm char #. Step 6.Report “operation successful” if user operation properly.
9	Step 1. On the phone, dial in ******** to start IVR Step 2.Choose “9”, and C3VOIP-150 report “ WEB Access Port” Step 3.Prompt “Please enter password”, the method of inputting password is same as operation 1. Step 4.Report “operation successful” if user operation properly. Step 5.Report the current WEB Access Port Step 6.Set the new WEB access port and with end char # Step 7. Report “operation successful” if user operation properly.
0	Step 1. On the phone, dial in ******** to start IVR Step 2.Choose “0”, and C3VOIP-150 report current Firmware version

4.4 Notice

- Under **Voice** menu, press *(star) to return to the top of the menu.
- If any changes made in the IP assignment mode, please reboot the C3VOIP-150 for the settings to take effect.
- When entering an IP address or subnet mask, input "*" after an address field and add "#" to complete the input
  For example, to enter the IP address 192.168.1.11 by keypad, press these keys: 192*168*1*11#.
- You can enter the password via phone keypad. The matching table between number and letters as follows:
  - To input: A, B, C, a, b, c -- press '2'
  - To input: D, E, F, d, e, f -- press '3'
  - To input: G, H, I, g, h, i -- press '4'
  - To input: J, K, L, j, k, l -- press '5'
  - To input: M, N, O, m, n, o -- press '6'
  - To input: P, Q, R, S, p, q, r, s -- press '7'
  - To input: T, U, V, t, u, v -- press '8'
  - To input: W, X, Y, Z, w, x, y, z -- press '9'
  - To input all other characters in the administrator password-----press '0',
    E.g. password is 'admin-admin', press '23646023646'.
- Press # (pound) key to indicate that you have finished entering the IP address or subnet mask or other settings.
- When assigning IP address in Static IP mode, customer must set IP address, subnet mask and default gateway. If in DHCP mode, please make sure that DHCP Server is available in your existing broadband connection to which WAN port of C3VOIP-150 is connected.

5. Parameter Introduction

5.1 Password

- There are 2 levels to access C3VOIP-150: **Administrator** level and **User** level. Passwords for these 2 levels are different.
- User at **Administrator** level can browse and set all configuration parameters, while user at **User** level can set all configuration parameters except SIP1/2 and some parameters that cannot be changed, such as server address and port. User has different access level with different password.
  - ✧ Default user with **Administrator** level：
    Username:admin, Password： admin
  - ✧ Default user with **User** level
    Username:admin, Password: user

5.2 URL format

The WEB login URL format is http://xxx.xxx.xxx.xxx., where "xxx.xxx.xxx.xxx" stands for the IP address of LAN or WAN port.

Below are two examples about the URL of LAN port and WAN port.

- **LAN port:**

Default URL of LAN port is: http://192.168.1.1

Note: 192.168.1.1 is C3VOIP-150 default LAN port's IP address

- **WAN port:**

Get WAN port address from IVR function or in Status/Basic webpage

(Assuming the IP is: 192.168.100.18)

The URL: HTTP://192.168.100.18

6. Login to the WEB Server

C3VOIP-150 has an embedded Web server that responds to HTTP get/post requests. User can use a Web browser like Internet Explorer to login and then configure C3VOIP-150.

6.1 Login WEB via the LAN port

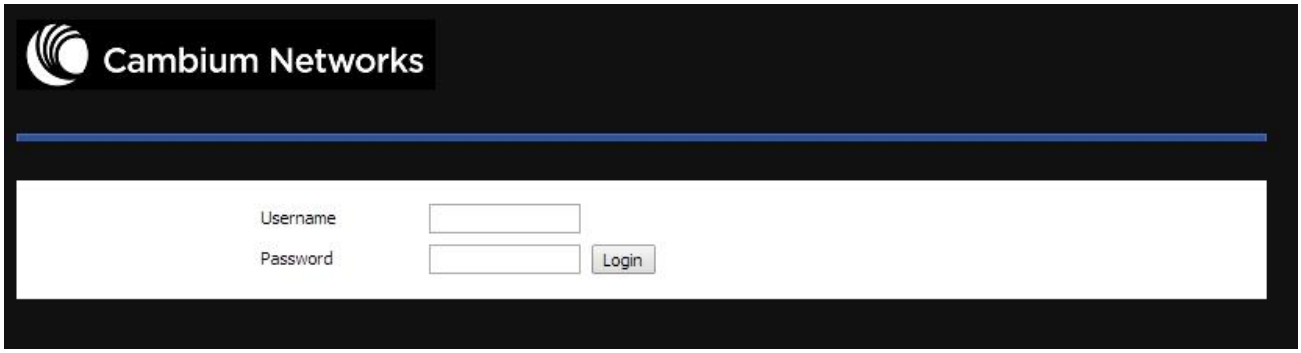
Step 1: Open WEB browser;

Step 2: Input the LAN port URL. Default is HTTP://192.168.1.1;

Note:

- ✧ User PC has the IP Address which is in the same segment of LAN port IP address, otherwise you cannot open the login page successfully.

Step 3: Once the right HTTP request is entered and sent by the Web browse， the ATA responds with the following login page.



Step 4: Input the password

Note: The password is case sensitive.

Step 5: First page user sees is the **Status** page.

6.2 Login WEB via the WAN port

Step 1: Open WEB browser;

Step 2: Lookup WAN port IP Address from IVR function or from WEB;

Step 3: Input the WAN port URL

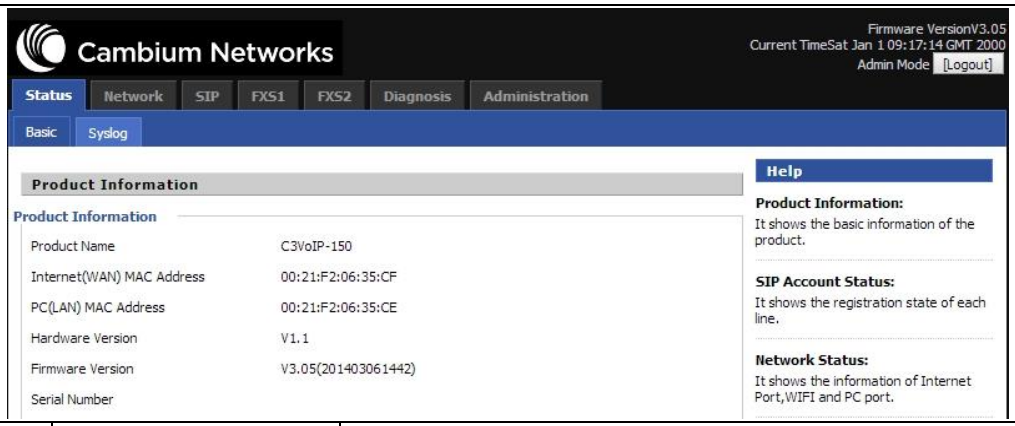
User's PC must have the IP Address which is in the same segment of WAN port IP address. Else you cannot open the login page.

Step 4: Once the right HTTP request is sent by the Web browser，the ATA responds with the login page.

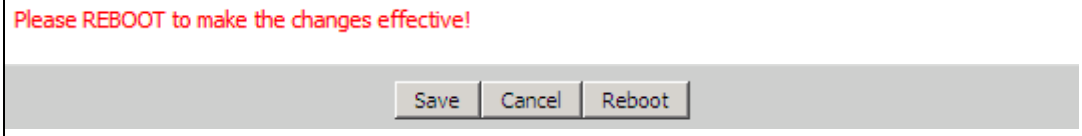
Step 5: Key in the password

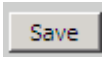
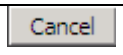
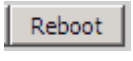
Note:
The password is case sensitive.

6.3 WEB Interface Introduction



	Name	Description
1	navigation bar	Click navigation bar. Few sub-navigation bars appear in the place 2
2	sub-navigation bar	Click sub-navigation bar to enter the configuration page
3	configuration title	The configuration title
4	configuration bars	The configuration bars
5	main information	Display the firmware version, DSP version, Current Time, and user can change login level (mode) to return to login page by press blue Switch button.
6	Help	Display the main information for configuration; user can get help from it directly.



	Every time user makes any changes, the button must be pressed to confirm the changes. ◆ After pressing the button, the red notice appears and prompts the user to 'Reboot', as shown below: Please REBOOT to make the changes effective!
	To cancel the changes.
	Press it to reboot C3VOIP-150.

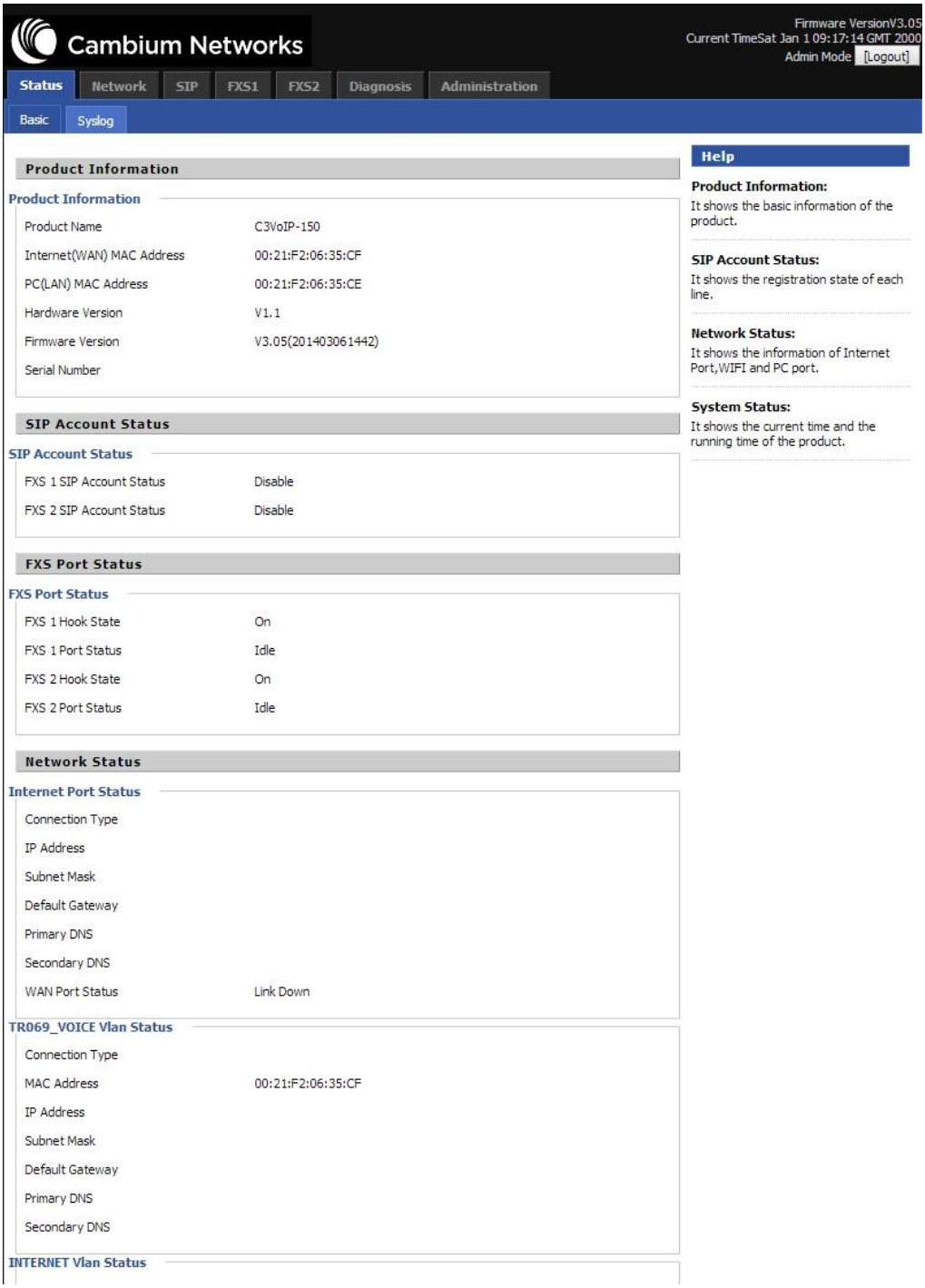
7. Configuration from the WEB page

7.1 Status

User can view C3VOIP-150 Basic and Syslog. It is the first page which user sees after a Web login.

7.1.1 Basic

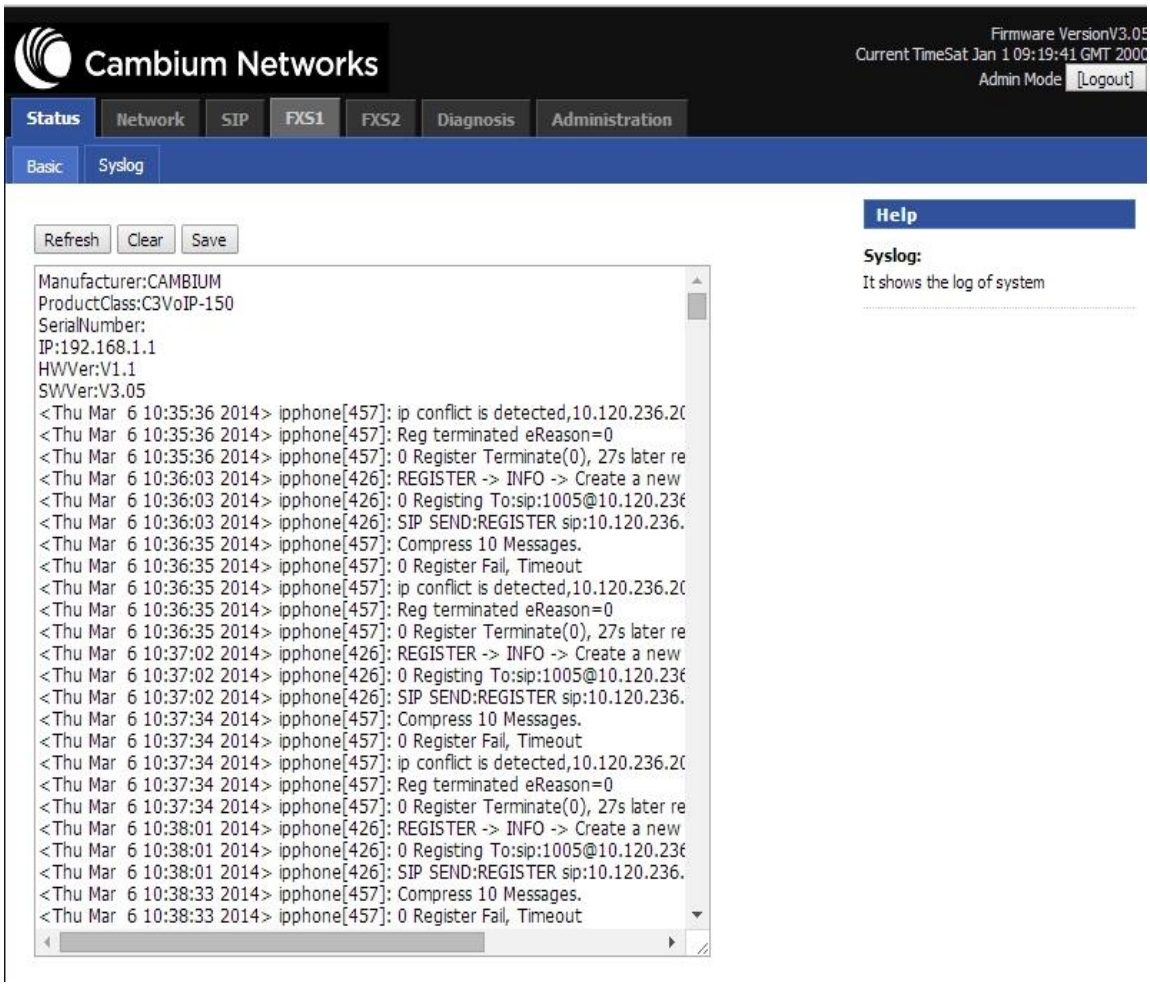
User can see the Product Information, SIP Account Status, Net Status, VPN Status, LAN Status, and System Status.



7.1.2 Syslog

In this configuration Interface user can view Syslog, which records the C3VOIP-150's important configuration information.

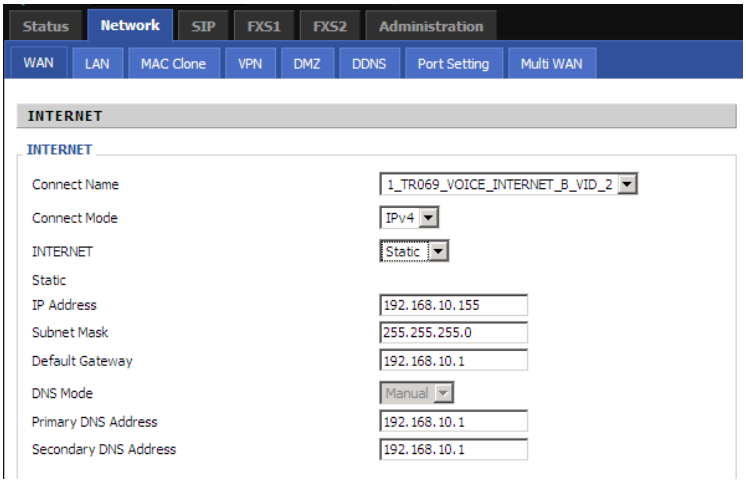
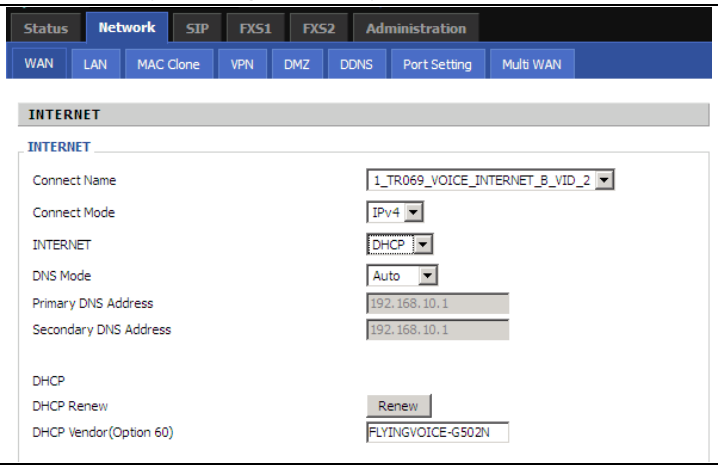
By default, syslog is enabled and the level is INFO, there are two system log level, INFO and Debug, in Debug level, there is more information. If you want to change, please go to **Administration/Management** page, **System Log Setting** column to change.

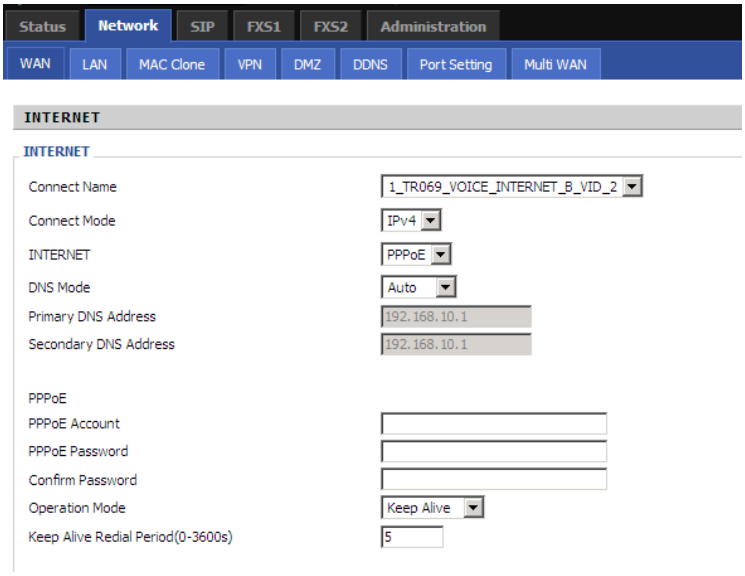


Click **Refresh** to reload the system log and click **Clear** to empty the current system log. To save the system log to your local PC, click **Save**.

7.2 Network

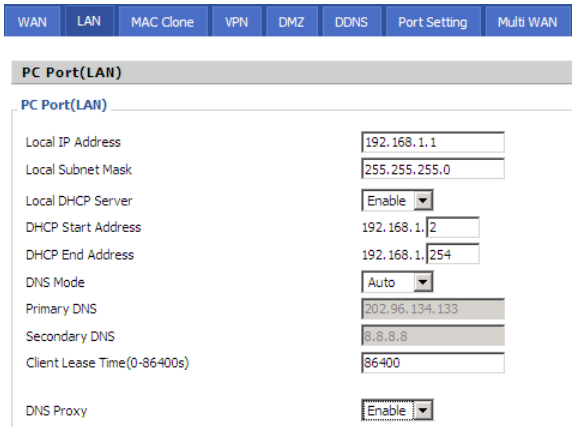
7.2.1 RADIO

<table>
<tr><td>Introduction</td><td colspan="2">The Internet Setup is to set WAN port mode, IP address and so on.
User can choose one WAN mode from Static, DHCP and PPPoE
Static: Users need to set IP Address, Subnet Mask, Gateway IP and DNS.
DHCP: C3VOIP-150 auto-configures the WAN parameter with immediate effect.
PPPoE: Users can enable C3VOIP-150 to connect to Internet by ADSL.</td></tr>
<tr><td rowspan="2">1.Static</td><td>WEB Interface</td><td> 
</td></tr>
<tr><td>Settings Introduction</td><td>1) Set “Static” in the “INTERNET” text.
2) Set IP address, the IP address is the one of the local area network.
3) Set Subnet Mask, it is usually “255.255.255.0” for the local area network.
4) Set Gateway, you can get it from your Administrator.
5) Set DNS, you can get it from your Administrator.</td></tr>
<tr><td rowspan="2">2.DHCP</td><td>WEB Interface</td><td> 
</td></tr>
<tr><td>Settings Introduction</td><td>1) Set “DHCP” in the “INTERNET” text.
2) DNS type: Manual and Auto
◆ In Manual: user must set the Primary DNS and Second DNS</td></tr>
</table>

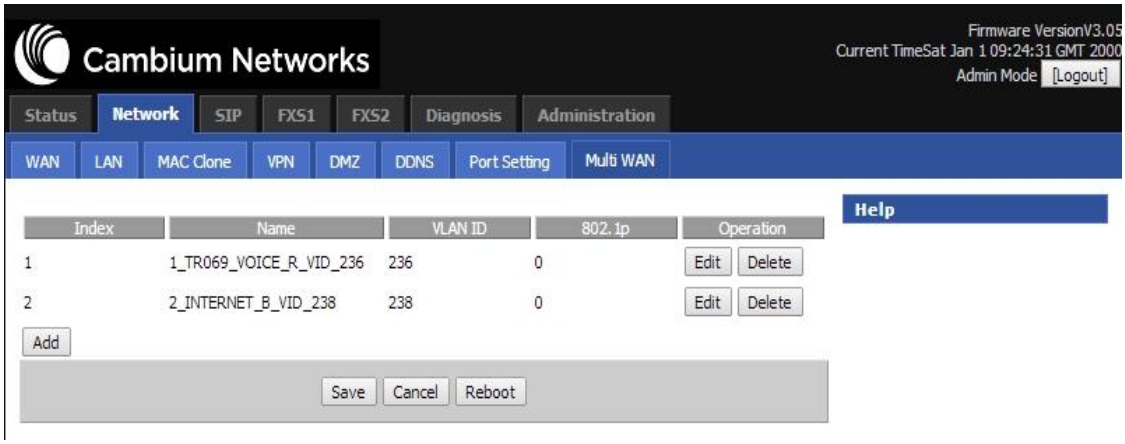
		manually. ◆ In Auto: C3VOIP-150 gets the Primary DNS and Second DNS from DHCP Server automatically.
3.PPPoE	WEB Interface	
	Settings Introduction	1) Set “PPPoE” in the “INTERNET” text 2) Fill the PPPoE account and password in the texts. 3) You must set “manual” in the “DNS Mode” if you set “DNS” by yourself. And then fill the DNS in the two following texts. Generally speaking, you can set “Auto” in the “DNS Mode” and C3VOIP-150 gets “DNS” from DHCP Server automatically. 4) You must click the “Reboot” in the left of the page to reboot the C3VOIP-150 if you see the words “Please REBOOT to make the changes effective!” After Reboot, you can see PPPoE Status and the network parameters in the System Status page, if C3VOIP-150 has connected successfully. You can select the “PPPoE” IP Mode if you are family users or your PC connects to Internet by ADSL. You must connect your PC with C3VOIP-150’s LAN. In detail, you can see the following LAN settings.
Port Bind		WAN connection cannot be shared between the binding port and finally bound port WAN connections bind operation washes away before the other WAN connection to the port binding operation! When you set several different VLANs for WAN port, user can use this option to define the LAN port’s service, usually please bind Internet VLAN to LAN port.

7.2.2 LAN

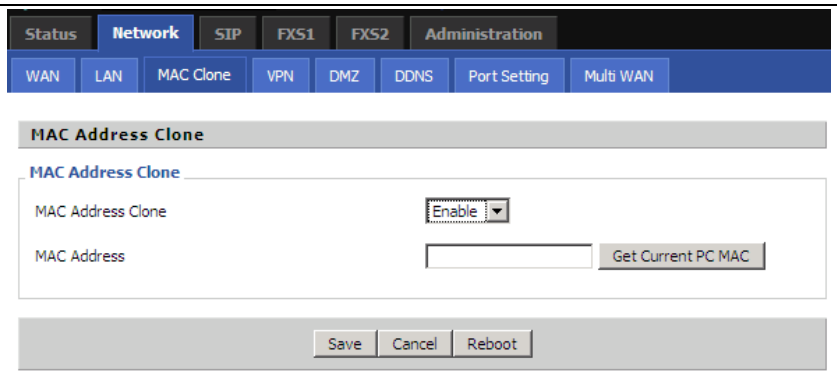
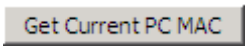
In this page, user can set the IP address of the device and set DHCP server. DHCP server is a kind of network function, C3VOIP-150 can supply DHCP service for the network which is linked with C3VOIP-150's LAN if you enable the DHCP server for C3VOIP-150's LAN (this is enabled by default).

WEB Interface	Settings Introduction
	1) Local IP Address and Local Subnet Mask Set the LAN port IP address and subnet mask. By default the address is 192.168.1.1 and subnet mask is 255.255.255.0. 2) Local DHCP Server and DHCP Start/End Address If or not enable the DHCP service, and set the start and end address of the DHCP server. By default DHCP server is enabled. 3) DNS Mode and Primary/Secondary DNS Set the DNS mode and address for the DHCP server. If set to Auto, the device gets the address from the upstream route, if set Manual, user need enter the primary and secondary DNS. 4) Client Lease Time Set client lease time. 5) DNS Proxy If or not enable DNS proxy. User can use the default settings.

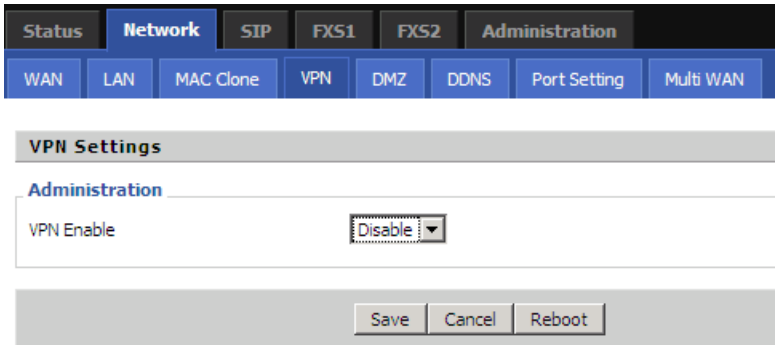
By default, the LAN port works in NAT mode, if user wants to make it work in Bridge mode, please go to **Network-->Multi WAN** page to set.



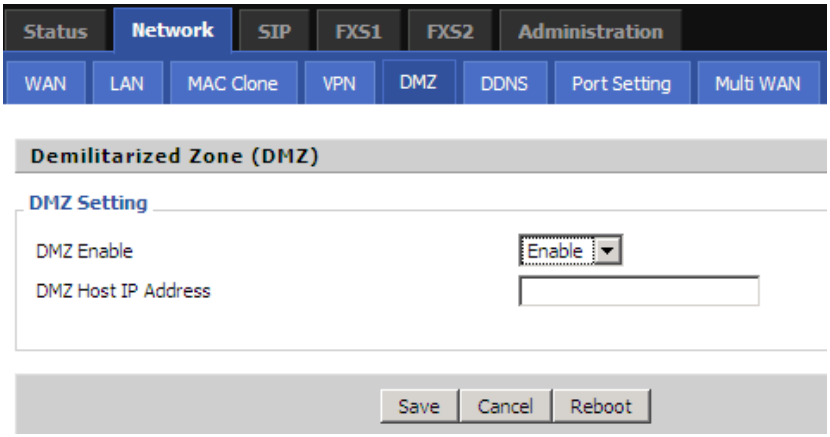
7.2.3 MAC Address Clone

Generally Introduction	MAC is the hardware address of network equipment. Sometimes, network providers may bind network accounts with the network equipment's MAC address. So you may not pass the provider's authentication when you use a new C3VOIP-150. In this case, you can use MAC Clone to copy your PC's MAC address to C3VOIP-150's WAN. MAC is an important parameter for network equipment, so ensure that the MAC is right, do not make C3VOIP-150 unusable. You can login C3VOIP-150's Web via LAN if you are incautious to make it wrong. And then cloning the right MAC or resume the default settings.
WEB Interface	
Settings Introduction	1). Press  button to get PC MAC address 2). Press to save the changes 3). Press Reboot to reboot C3VOIP-150

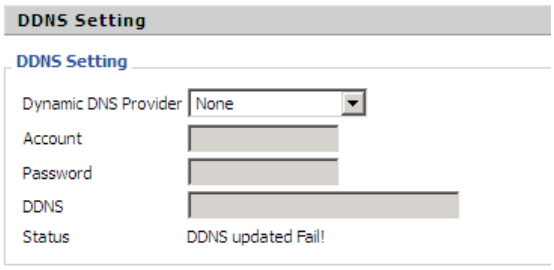
7.2.4 VPN Settings

Generally Introduction	C3VOIP-150 has two kinds VPN: PPTP and L2TP.
WEB Interface	
Settings Introduction	1) VPN Enable: If or not enable VPN, user can select from PPTP and L2TP. 2) Initial Service IP：VPN server IP address 3) User Name: the user name for authentication 4) Password: password for authentication

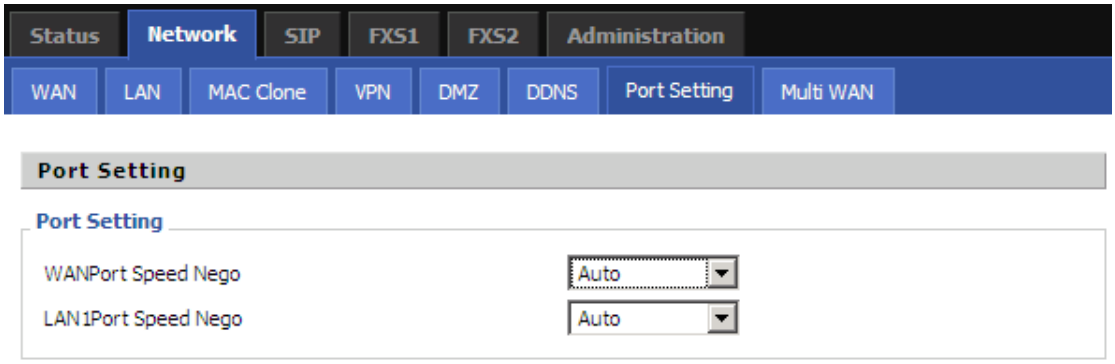
7.2.5 DMZ

Generally Introduction	◆ C3VOIP-150 rejects the outside requests if you enable the NAT. However, sometimes it is needed to access the PC which is linked with C3VOIP-150's LAN to use the PC's service. Now, you must use the C3VOIP-150's DMZ to realize it. ◆ Here, DMZ is the same with mapping ports for network equipment. ◆ PC which is linked with C3VOIP-150's LAN can get the requests from the LAN by some ports of C3VOIP-150's WAN retransmits. (Your PC is DMZ computer for short as follows.) ◆ User must enable "NAT" mode when want to use DMZ.
WEB Interface	
Settings Introduction	1) DMZ Enable: if or not enable DMZ 2) DMZ Host IP Address: set the IP address of DMZ host For example, the DMZ computer's IP is "192.168.1.2", the DMZ function is that DMZ computer can get the requests from the ports of C3VOIP-150's WAN port.

7.2.6 DDNS

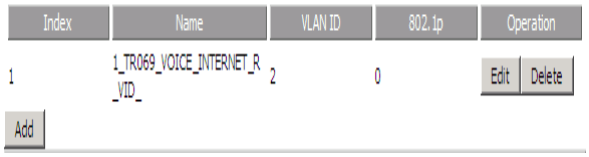
Webpage	Description
	Dynamic DNS provider: Select the DDNS provider.
	Account: Enter the account user gets from the DDNS provider.
	Password: Enter the password user gets from the DDNS provider.
	DDNS: Enter the DDNS domain.
	Status: Displays the DDNS connection status.

7.2.7 Port Setting



In this webpage, user can set WAN/LAN port speed negotiation, default are Auto and options are 100 Mbps Full, 100 Mbps Half, 10 Mbps Full and 10 Mbps Half.

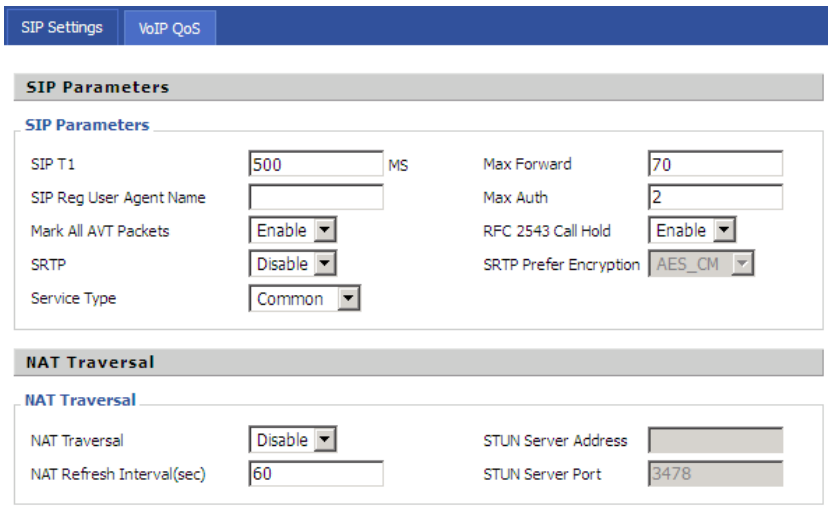
7.2.8 Multi WAN

Webpage	Description
 Picture 1	Name: The name of VLAN. Edit: Click **Edit** to change the current VLAN. Delete: Click **Delete** to remove the current VLAN. Add: Click **Add** to insert another VLAN.
When you click **Edit**, the webpage turns out to Picture 2.	
Picture 2	VLAN ID: Set the VLAN ID, you can choose from 2 to 4094. 802.1p: 802.1p priority, user can choose from 0 to 7. Bridge Mode: Set network connection mode, it can be Route or Bridge. Service: Select the service mode of this port. Take VLAN Tag: If or not enable take VLAN tag.

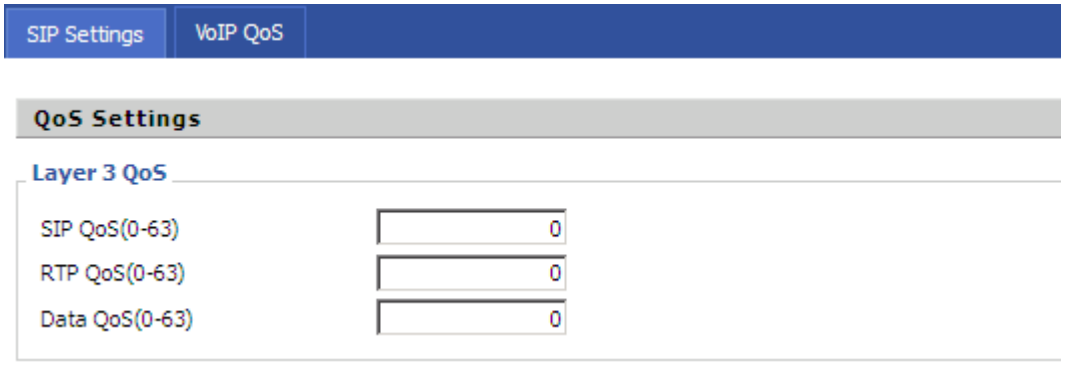
For further details on multi WAN, refer the *Multi WAN User Manual*.

7.3 SIP

7.3.1 SIP Settings

WEB Interface	
Settings Introduction	1) SIP Parameters: This page displays the SIP parameters. 2) NAT Traversal If you want traverse NAT/Firewall, please enable NAT Traversal and fill in the corresponding parameters.

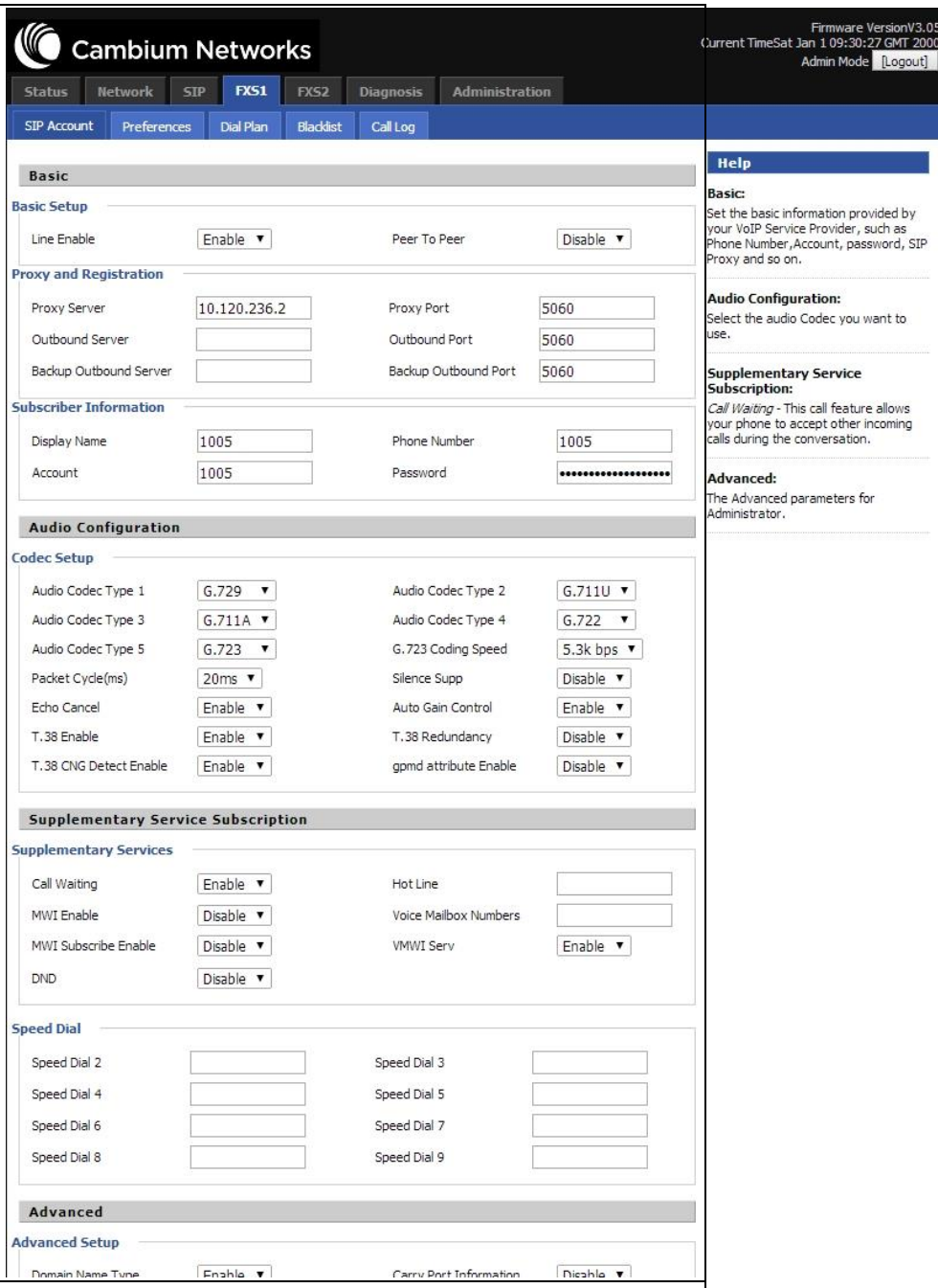
7.3.2 VoIP QoS

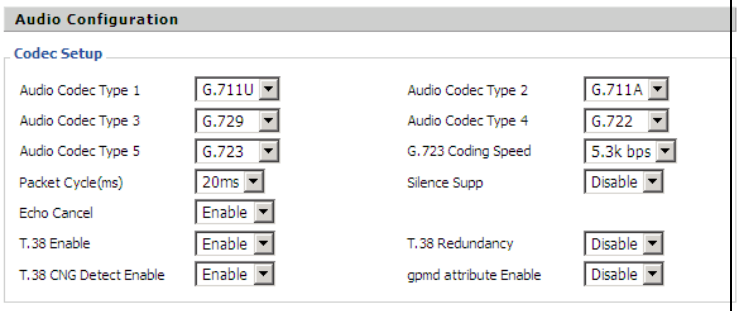


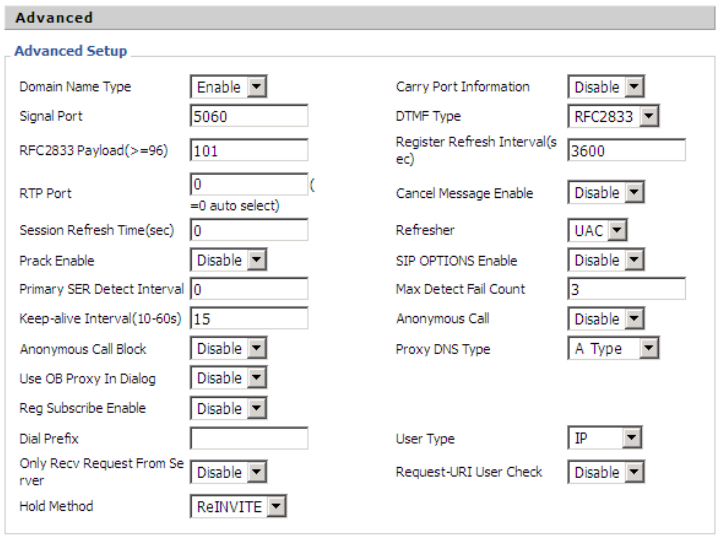
Some ISPs supply QoS services. The QoS services can make the best of improving the quality of Voice application. You can get the settings from the ISP if they supply QoS services. Please contact them if you need it.

7.4 FXS1

7.4.1 SIP Account

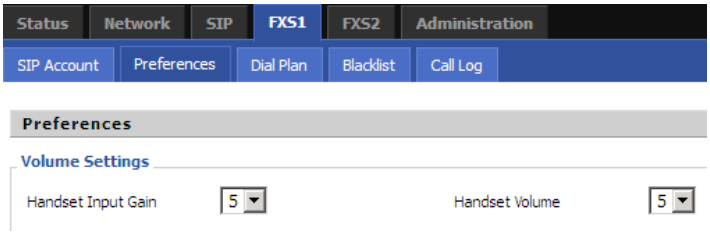
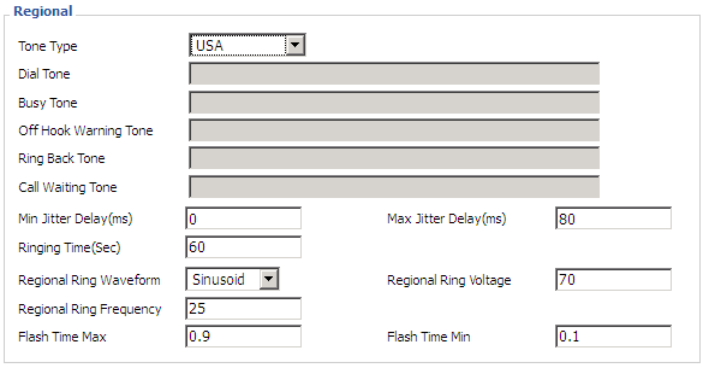
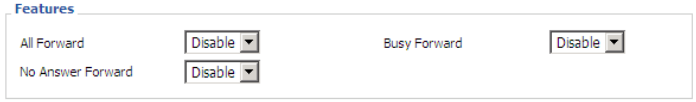
<table>
<tr>
<td rowspan="2">Basic</td>
<td>WEB Interface</td>
<td> 
</td>
</tr>
<tr>
<td>Settings Introduction</td>
<td>1) Line Enable: To enable FXS1
2) Peer to Peer: To enable Peer To Peer on FXS1 port.
3) SIP Proxy: The IP address of SIP Server.
4) SIP Proxy Port: The port which SIP Server supports for VOIP service, default is 5060.
You must enable “Carry Port Information” in the Other Settings page if the SIP Server Port is not 5060 or SIP messages need to carry port information.</td>
</tr>
</table>

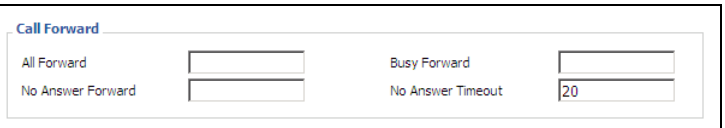
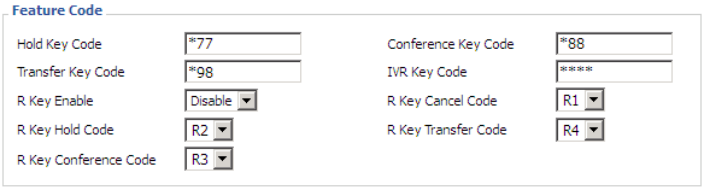
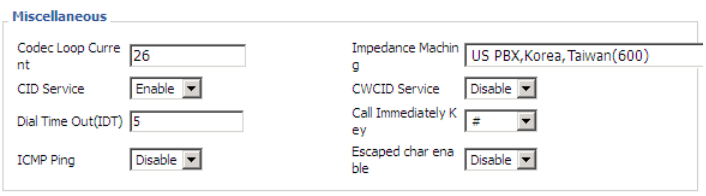
<table>
<tr><td></td><td></td><td>5) Outbound Proxy: outbound Proxy ip or domain name.
6) Outbound Proxy Port: outbound Proxy’s Service port.
7) Backup Outbound Proxy: backup outbound proxy IP or Domain name.
8)Backup Outbound Port: backup outbound Proxy’s Service port
9) Display Name: The number is display in caller field
10) Phone Number: Number of telephone provided by SIP Proxy.
11) Account: Account of telephone provided by SIP Proxy.
12) Password: Enter the password of the account.</td></tr>
<tr><td rowspan="2">Audio Configuration</td><td>WEB Interface</td><td> 
</td></tr>
<tr><td>Settings Introduction</td><td>1) Audio Codec: There are 5 kinds of Audio Coding Modes: G.711A, G.711U, G.722, G.729 and G.723. And G.723 support 5.3kbps and 6.3kbps coding speed.
2) Packet Cycle (ms): the RTP packet cycle time
3) Silence Supp Enable: if or not enable silence
4) Echo Cancel: if or not enable echo cancel
5) T.38 Enable: if or not enable T.38
6) T.38 Redundancy: if or not enable T.38 redundancy.
7) T.38 CNG Detect Enable: if or not enable CNG detect.
8) GPMD Attribute Enable: if or not enable GPMD attribute.</td></tr>
<tr><td rowspan="2">Supplementary Service Subscription</td><td>WEB Interface</td><td>
</td></tr>
<tr><td>Settings Introduction</td><td>1) Call Waiting: By default it is enabled.
2) Hot Line: Enter the hot line number, after user configure this, the device dials out the hot line number automatically when user pick up the handset or press the speaker button.
If you want to delay some seconds, please add delay time with T behind the number, for example, 511T4, the device delays for about 4 seconds before dialing the number when you pick up the handset.
3) MWI Enable: To enable or disable MWI.</td></tr>
</table>

		4) Voice Mail box Numbers: enter the voice mail box number. 5) DND: If or not enable DND.
Advanced Setup	WEB Interface	
	Settings Introduction	1) Domain name Mode: Domain name in the SIP URI. 2) Carry Port Information: Port information in the SIP URI. 3) Signal Port: The local port of SIP protocol, default is **5060**. 4) DTMF Type: choose the DTMF type from IN_band, RFC2833 and SIP INFO. 5) RFC2833 Payload (>=96): User can use the default setting. 6) Register Refresh Interval (Second): The interval between two normal Register messages. You can use the default setting. 7) RTP Port: C3VOIP-150 selects idle port for RTP if you set "0", otherwise C3VOIP-150 use the value you set. Generally speaking, set "0". 8) Cancel Message Enable: when you set enable, an unregistered message is sent before registration, while you set disable, unregistered message is not sent before registration. You must set the option for different Proxy. 9) Session Refresh Time: Set the session refresh time. 10) Refresher: Select from UAC and UAS. 11) Prack Enable: To enable or disable prack. 12) SIP Options Enable: To enable or disable sip options. 13) Primary SER Detect Interval: Set the primary server detect interval. 14) Max Detect Fail Count: The maximum detect fail account. 15) Keep-alive interval (10-60s): the interval that we send an empty packet to Proxy. 16) Anonymous Call: To enable or disable anonymous call. 17) Anonymous Call Block: If or not enable block anonymous call function. 18) Proxy DNS Type: Select from A Type and DNS SRV. 19) Use OB Proxy In Dialog: To use OB proxy in dialog or not.

		20) Reg Subscribe Enable: To enable or disable registration subscribe. 21) Dial Prefix: Define the prefix of the phone number you dialed. 22) User Type: Define the user type, options are IP and Phone. 23) Only Recv Request From Server: If this option is enabled, C3VOIP-150 only receives call from the same SIP server. 24) Request URI user check: Enable or disable request URI user check. 25) Hold Method: Select the hold method, options are ReINVITE and INFO.

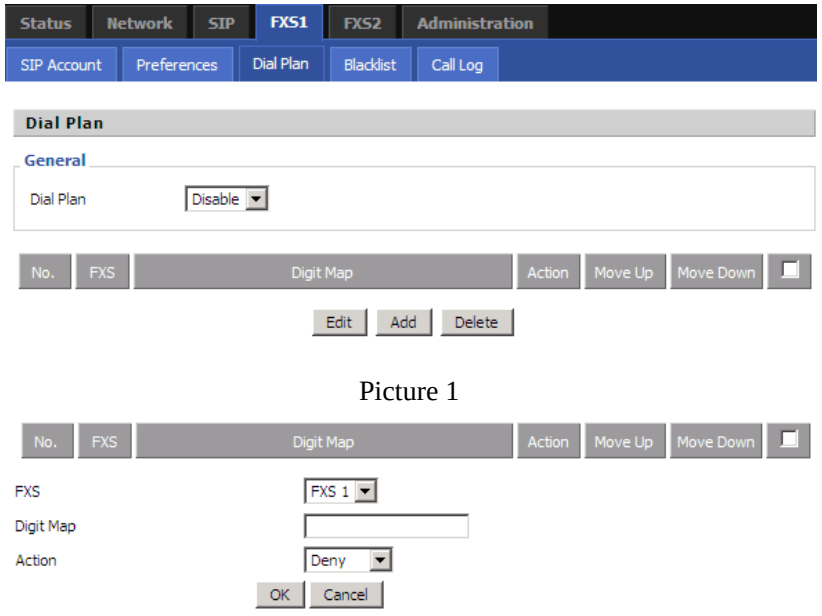
7.4.2 Preferences

Preferences	WEB Interface	
	Settings Introduction	1) Handset Input Gain: adjust the input gain from 0-7. 2) Handset volume: adjust the output volume from 0-7.
Regional	WEB Interface	
	Settings Introduction	In this page, user can define tone type and set some parameters of the FXS1 port.
Features	WEB Interface	
	Settings Introduction	All Forward: To enable or disable all call forward; Busy Forward: To enable or disable call forward when FXS1 port is busy; No Answer Forward: To enable or disable call forward when FXS1 port does not answer the call.

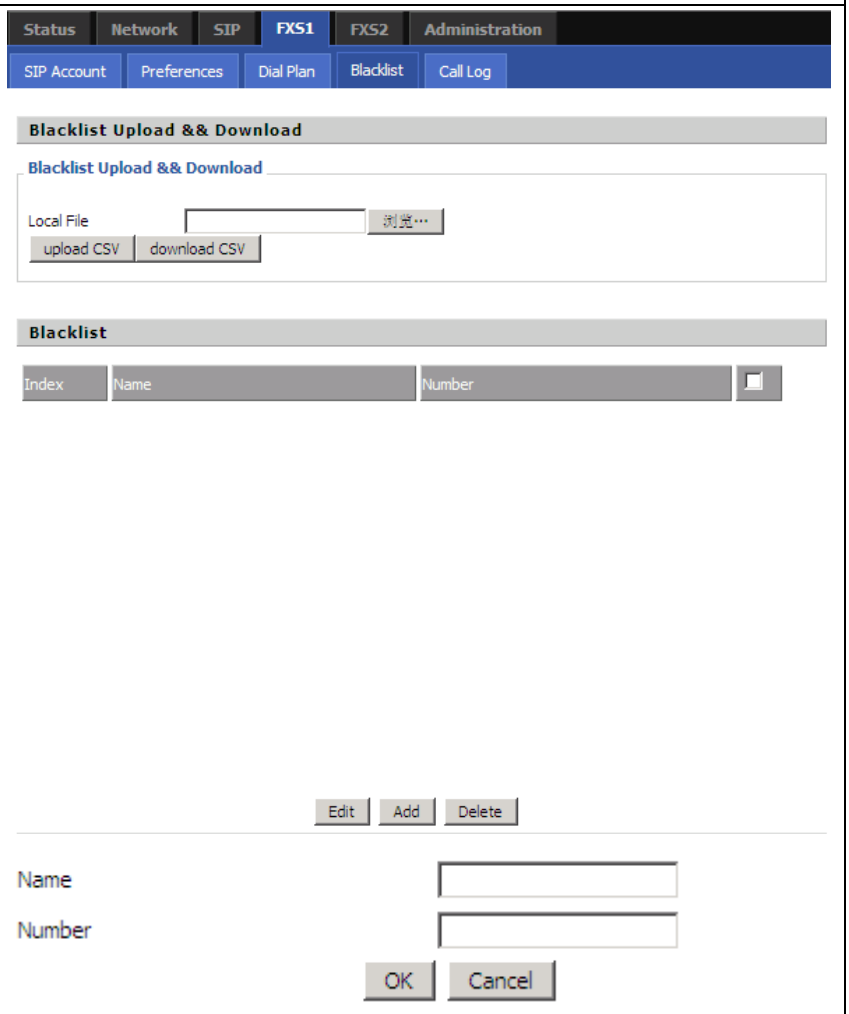
<table>
<tr><td rowspan="2">Call Forward</td><td>WEB Interface</td><td> 
</td></tr>
<tr><td>Settings Introduction</td><td>All Forward: Enter the all call forward destination number.
Busy Forward: Enter the busy call forward destination number.
No Answer Forward: Enter the “No answer forward” destination number.
No Answer Timeout: Define the “No answer timeout” time.</td></tr>
<tr><td rowspan="2">Feature Code</td><td>WEB Interface</td><td> 
</td></tr>
<tr><td>Settings Introduction</td><td>The feature code for call hold, conference, transfer and IVR. User can also use R key to perform these functions. For example, press R and then press 2 to hold the call.</td></tr>
<tr><td rowspan="2">Miscellaneous</td><td>WEB Interface</td><td> 
</td></tr>
<tr><td>Settings Introduction</td><td>Codec Loop Current: Set the value for codec loop current.
Impedance Matching: Set impedance value.
CID Service: If or not enable caller ID service.
CWCID Service: To enable or disable call waiting caller ID service.
Dial Time Out: Define how long the device waits before sending out the phone numbers.
Call Immediately Key: Add this key after the phone numbers, C3VOIP-150 dials in the numbers immediately.
ICMP Ping: To enable or disable ICMP ping.
Escaped char enable: To enable or disable escaped char.</td></tr>
</table>

7.4.3 Dial Plan

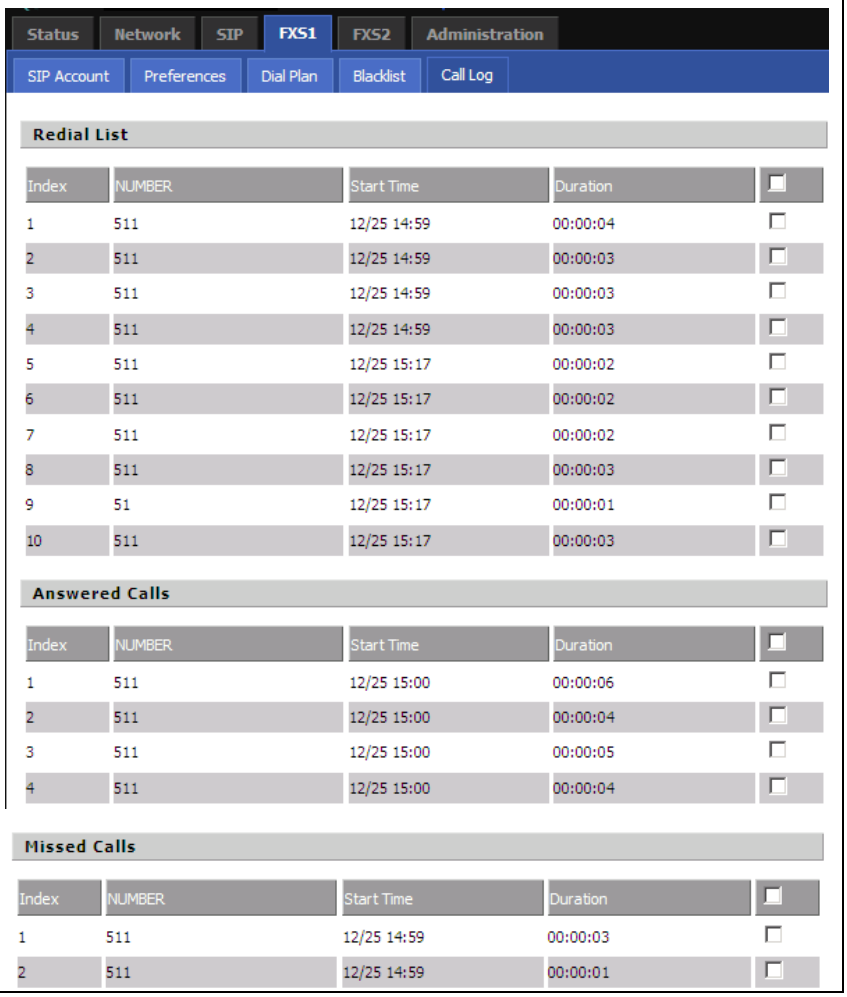
If you set a shortcut number for a dial rule and dial a shortcut number, C3VOIP-150 realizes the whole dial rule immediately.

WEB Interface	 Picture 1 Picture 2
Settings Introduction	Dial Plan: To enable or disable dial plan. Edit: To edit the dial rule user selected. Add: To add a new dial rule. Delete: To delete the dial rule user selected. Once you click **Add**, the webpage shows up as shown in **Picture 2**. FXS: Select which FXS port uses the rule. Digit Map: The digit map sting of dial plan, details please refer the *Dial Plan User Manual*. Action: Choose the action, when user's dial string matches the dial rule. The device's action is to deny or dial out.

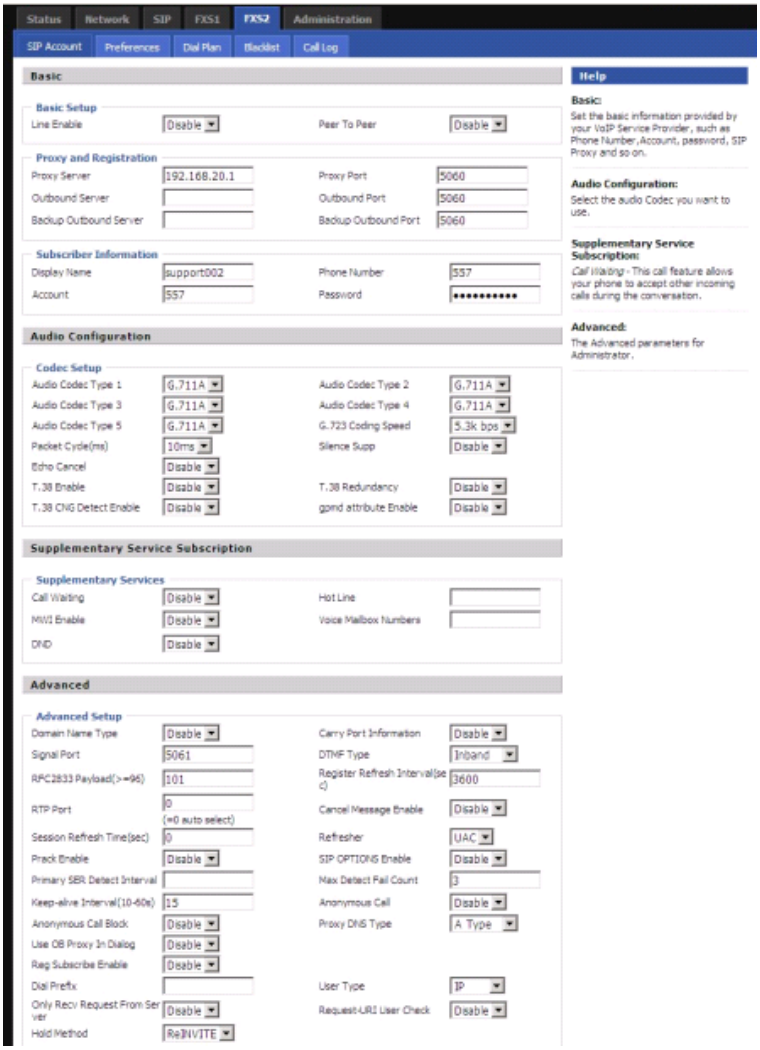
7.4.4 Black List

Settings Introduction	In this configuration interface, you can set the blacklist, enter the name and phone number you wish to block. User also can upload a piece of Blacklist file. If user sets a block on the number 2222, 2222 cannot call you, but you can call 2222. Enter the name and Number you want to add to the blacklist.
WEB Interface	

7.4.5 Call Log

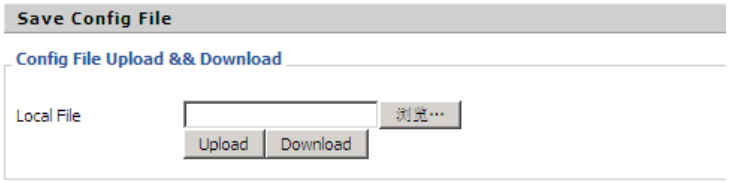
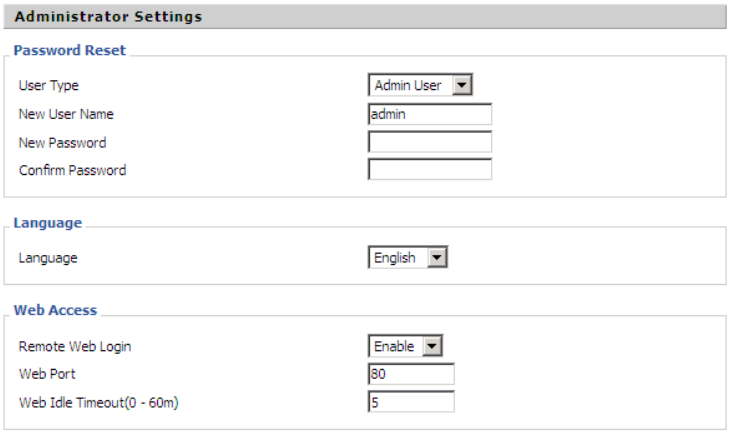
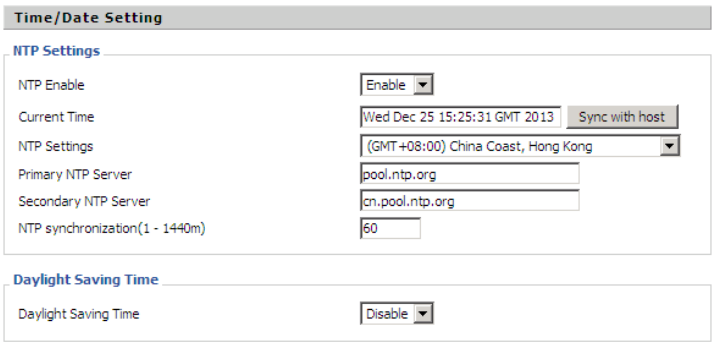
Settings Introduction	User can view C3VOIP-150 call log, including missed, dialed and answered calls. User can also delete the call logs.
WEB Interface	

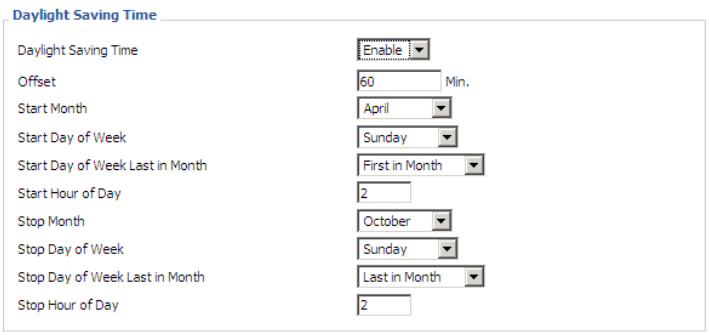
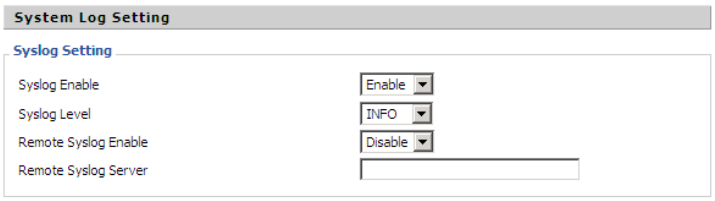
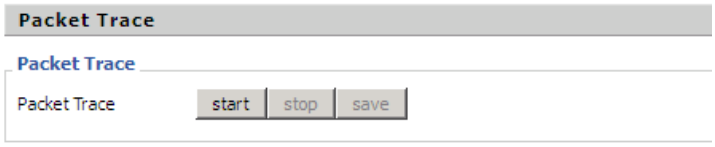
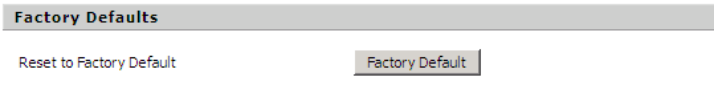
7.5 FXS2

Settings Introduction	The settings of FXS2 are the same as FXS1.
WEB Interface	

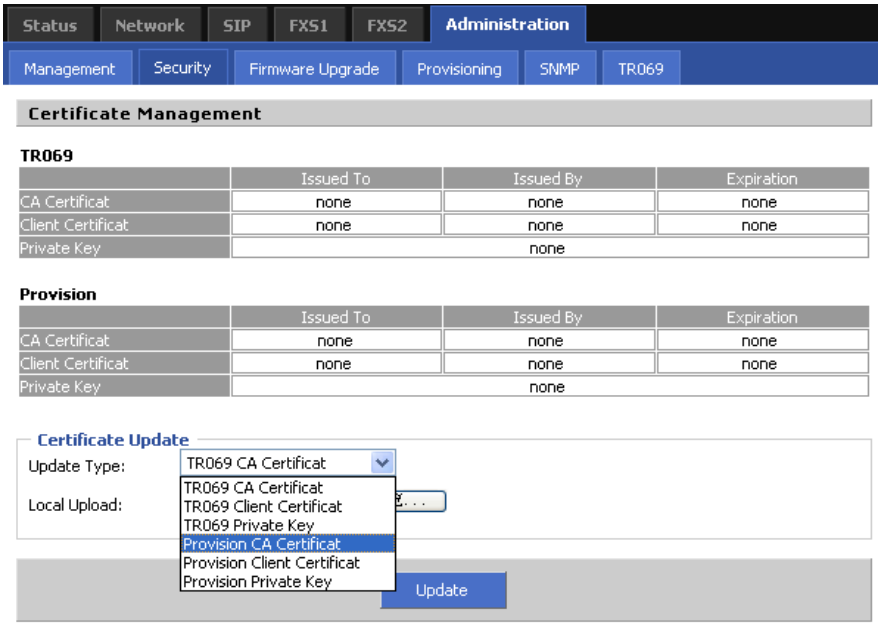
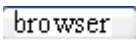
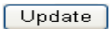
7.6 Administration

7.6.1 Management

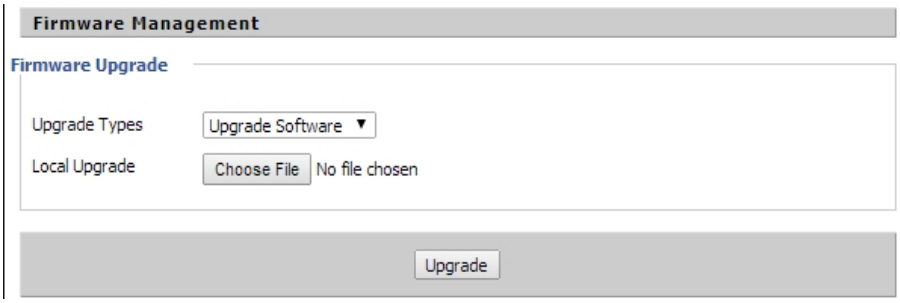
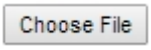
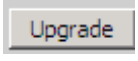
Save Config File	WEB Interface	
	Settings Introduction	In this column, user can upload a new configuration file for the device and also can save the configuration file to local PC.
Administrator Settings	WEB Interface	
	Settings Introduction	1) Password Reset: Reset username and password, user need select one user level first and then rename the user name and change the password. 2) Language: Change the language. C3VOIP-150 supports English, Russian, Finnish, Spanish, Chinese, etc. 3) Web Access: Enable or disable web access, user can also define the web access port and web idle timeout.
Time/Date	WEB Interface	

		
	Settings Introduction	1) NTP Settings: User can enable the NTP and select time zone and set NTP server and so on. 2) Daylight Saving Time: To enable or disable daylight saving time, this is the function to bring an hour ahead the normal time. When this option is enabled, user can define offset, start/stop month, start/stop day of week, start/stop day of week and start/stop hour of day.
System Log Setting	WEB Interface	
	Settings Introduction	1) Syslog Enable: To enable or disable system log. 2) Syslog Level: C3VOIP-150 has two log levels, INFO and Debug. In Debug level, more information is in the system log. 3) Remote Syslog Enable: To enable or disable remote system log. 4) Remote Syslog Server: Enter the IP address of the remote system log server, C3VOIP-150 sends the system log to the host. To check system log, user must open syslog server on your local PC, tftp32 can be used as system log server.
Packet Trace	WEB Interface	
	Settings Introduction	Press **Start** to begin tracing the packet or press **Stop** to cancel. Also to save the packets to your local PC, click **Save**.
Factory Default	WEB Interface	
	Settings Introduction	Press the **Factory Default** button to load the default settings on the device.

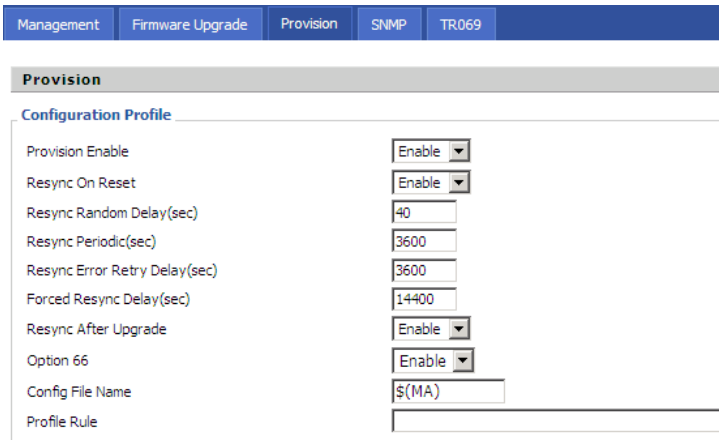
7.6.2 Security

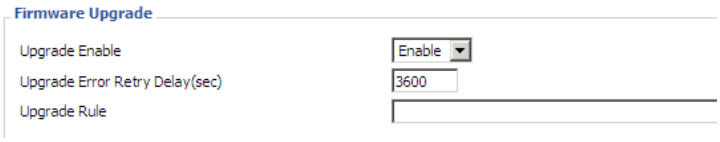
On this page, user can upload HTTPs provision SSL certificates or upload the TR069 certificates. If C3VOIP-150 deletes the HTTPs provision and this webpage, Flyingvoice adds the HTTPs provision soon in the new firmware, so we still keep this section.	
WEB Interface	
Settings Introduction	User can upload certain files for TR069 and Provision. Steps: 1) Choose File Types in 2) Press  to browser file. 3) Press  to start upgrading.

7.6.3 Firmware Upgrade

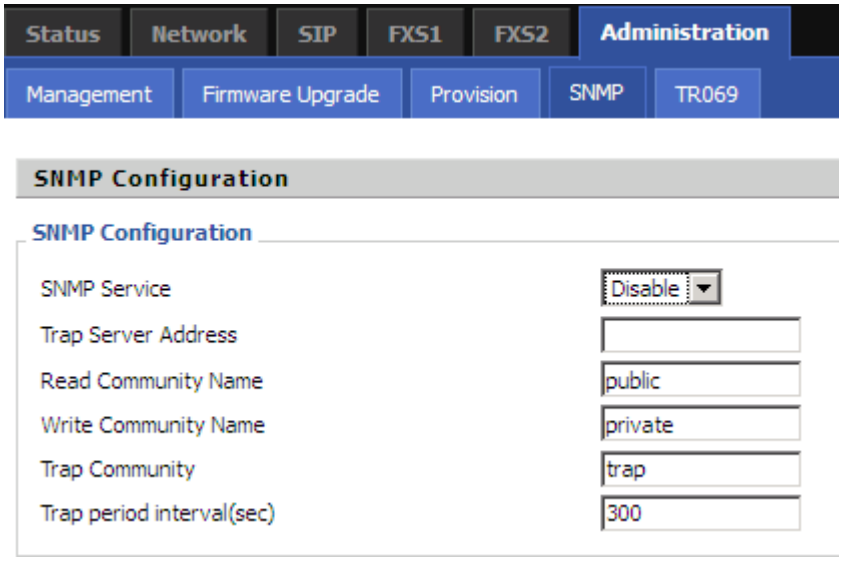
WEB Interface	
Settings Introduction	1) Press  to select a firmware file. 2) Press  to start upgrading.

7.6.4 Provisioning

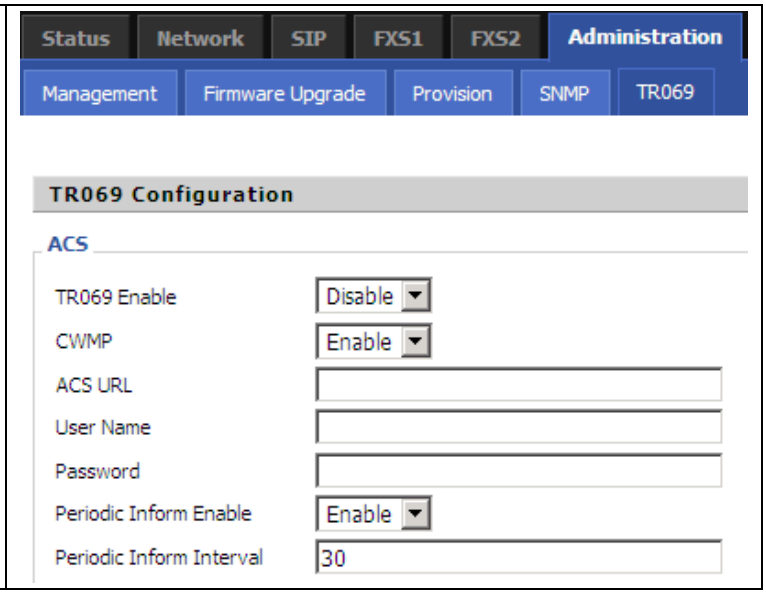
<table>
<tr><td>Generally Introduction</td><td colspan="2">1) Provisioning allow C3VOIP-150 auto-upgrading or auto-configuring
2) The current C3VOIP-150 supports 2 ways to provision: TFTP and HTTP.
◆ Before testing or using TFTP, user must have TFTP server and upgrading file and configuring file.
◆ Before testing or using HTTP, user must have HTTP server and upgrading file and configuring file.
◆ Before testing or using HTTPS, user must have HTTPS server, upgrading file and configuring file and CA Certificate file (must be same as HTTPS server's) and Client Certificate file and Private key file (HTTPs provision will be supported soon)
User can uploading CA Certificate file and Client Certificate file and Private Key file in Security page.
Notice: Please refer to documentation Provision_User Manual_en_v1.5.pdf to use this function.</td></tr>
<tr><td rowspan="2">Configuration Profile</td><td>WEB Interface</td><td> 
</td></tr>
<tr><td>Settings Introduction</td><td>1) Provision Enabled: To enable or disable provision
2) Re-sync On Reset: To enable or disable re-sync after DIV378 restart
3) Re-sync Random Delay: Set the maximum delay for request the synchronization file, default is 40.
4) Re-sync Periodic: Set the periodic time for re-sync, default is 3600s.
5) Re-sync Error Retry Delay: If the last re-sync failed, C3VOIP-150 tries a re-sync after the "Re-sync Error Retry Delay" time. The default is 3600s.
6) Forced Re-sync Delay: If it's time to re-sync, but C3VOIP-150 is busy. In this case, C3VOIP-150 waits for a period time, the longest is "Forced Re-sync Delay", the default is 14400s. When the time exceeds, C3VOIP-150 is forced to re-sync.
7) Re-sync After Upgrade Attempt: To enable or disable firmware upgrade after re-sync. The default is enabled.</td></tr>
</table>

		8) Option 66: To enable or disable DHCP option 66 to override server. If enable, the parameter "profile rule" has no effect. 9) Config File Name: It is used for In-house provision mode only. When use TFTP with option 66 to realize provisioning, user must input right configuration file name in IP542N's webpage. When disable **Option 66**, this parameter has no effect. 10) Profile Rule: URL of profile provision file Note that the specified file path is relative to the TFTP server's virtual root directory.
Firmware Upgrade	WEB Interface	
	Settings Introduction	1) Upgrade Enable: To enable or disable firmware upgrade via provision. 2) Upgrade Error Retry Delay: If the last upgrade fails, C3VOIP-150 tries upgrading again after "Upgrade Error Retry Delay" period, default is 3600s. 3) Upgrade Rule: URL for the upgrade file

7.6.5 SNMP

WEB Interface	
Settings Introduction	1) SNMP Enable: To enable or disable SNMP 2) Trap Server Address: Enter the trap server address. 3) Read Community Name: string, as an express password between management progress and agent progress. 4) Write Community Name: String, as an express password between management progress and agent progress. 5) Trap Community: The community field in trap. 6) Trap Period interval: The interval for sending the trap.

7.6.6 TR069

TR069 Configuration	WEB Interface	

	Settings Introduction	1) TR069 Enable: To enable TR069 or not. 2) CWMP: If or not enable TR069 3) ACS URL: The URL of TR069 server. 4) User Name: Enter the user name to connect to TR069 server. 5) Password: Enter the password to connect the TR069 server. 6) Periodic Inform Enable: If or not enable periodic information. 7) Periodic Inform Interval: The interval of sending information to TR069 server.
Connect Request	WEB Interface	
	Settings Introduction	1) User Name: Define the username for the TR069 server connecting C3VOIP-150. 2) Password: Define the password for TR069 server connecting C3VOIP-150. 3) SSL Key: Define the SSL key for TR069 server connecting to C3VOIP-150.

8. Functions

8.1 Making Calls

- Dial the number directly and wait for 5 seconds (by default, it has No Key Entry Timeout).
- Dial the number ending with char #. C3VOIP-150 dials out immediately
- Dial the phone number which matches one dial rule. C3VOIP-150 dials out immediately and there is no need to press # or wait for 5 seconds.

8.2 Call Waiting

Step 1.Enable waiting feature in **FXS1/FXS2→Preference→Supplementary Service→Call Waiting** (default is Enable)

Step 2.While in conversation, user hears a special stutter tone if there is another incoming call.

Step3.User then can press ***77** (or Flash button) to put the current call party on hold automatically and switch to the other call. Pressing ***77** toggles between two active calls.

8.3 Call Hold

Step 1.While in conversation, pressing the ***77** puts the remote end on hold.

Step 2.Pressing the ***77** again, releases the previous "Hold state" and resumes the bi-directional media.

8.4 Call Transferring

C3VOIP-150 supports blind transfer and attended transfer.

8.4.1 Blind Transfer

Assuming that call party A and party B are in conversation, A wants to Blind Transfer B to C

Step 1.Party A dials ***98**, A hears a dialing tone

Step 2.Dial party C's number, and press # (or wait for 5 seconds) to call C, then C rings, A is disconnected.

Step 3.If C answer the call, B and C can go on talking.

8.4.2 Attended Transfer

Assuming that call party A and B are in conversation. A wants to Attend Transfer B to C:

Step 1.Party A dials ***77** to hold B, A hears a dialing tone

Step 2.Dial party C's number, and press # (or wait for 5 seconds) to call C, then A hears ringing tone.

Step 3.If C answer the call, A talks to C first

Step 4.If C wants to talk with B, A press ***98** to transfer, and then C will talk with B. If C does not talk with C successfully, A will talk with B again.

8.5 3-way conference call

Assuming that call party A and B are in conversation. A wants to add C to the conference:

Step 1.Party A dials ***77** to hold B, A hears dial tone

Step 2.Dial party C's number, and press # (or wait for 5 seconds) to call C, then A hears ring tone.

Step 3.If C answers the call, A talks to C first

Step 4.If C receive the conference, A press ***88** to add C to the conference, and then A, B and C are in conference.

8.6 Call Forwarding

Step 1: Open **FXS1/FXS2→Preferences→Call Forward**, enable the one call forward mode and fill forwarded number

Step 2: C3VOIP-150 forwards incoming call to the forwarded number according to the settings of Call Forward and call status

8.7 Direct IP calls

Direct IP calling allows two phones, that is, an ATA with an analog phone and another VoIP Device, to talk to each other without a SIP proxy. VoIP calls can be made between two phones if:

- Both ATA and the other VoIP device (i.e. another ATA or other SIP products) have public IP addresses, or
- Both ATA and the other VoIP device (i.e. another ATA or other SIP products) are on the same LAN using private or public IP addresses, or
- Both ATA and the other VoIP device (i.e. another ATA or other SIP products) can be connected through a router using public or private IP addresses

To make a direct IP call,

Step 1: Picking up the analog phone or turning on the speaker phone on the analog phone

Step 2: Input the IP address directly with ending char #. E.g. call 192.168.20.34, dial 192*168*20*34#

8.8 Speed dialing

Notice: The current firmware does not support speed dial, will be added in new branch firmware.

Step 1: Set phone number (E.g.3333) in **FXS1/2→SIP Account→Speed Dial** and then save the changes

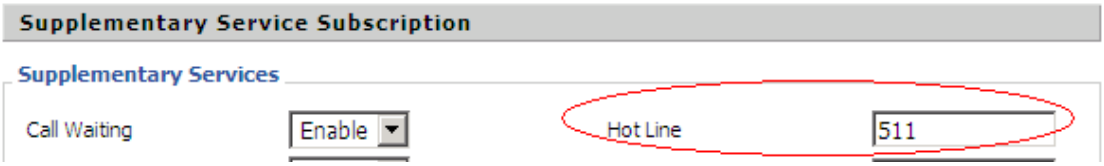
Step 2: Dial ***74** to active speed dial function

Step 3: Then dial **2** to call 3333, and C3VOIP-150 dials out immediately.

8.9 Hotline

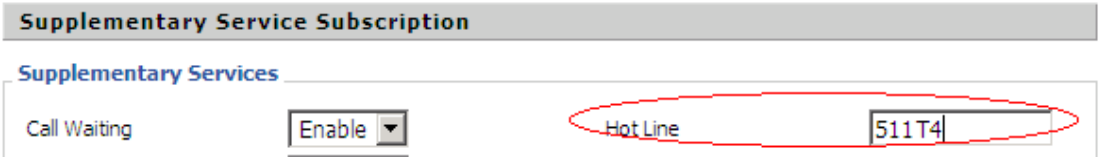
Step 1: Set hotline in **FXS1/2→SIP Account→Supplementary Service Subscription** (see Figure 3). And then save the changes.

Figure 3 Hotline



Step 2.Picking up handset or press speaker button, C3VOIP-150 will ring hotline immediately.

If you want to delay some seconds after pick up the handset, please add delay time. For example, C3VOIP-150 will call 511 after user picks up the handset for 4 seconds.



8.10 Daylight Saving Time

Daylight Saving Time (or summer time as it is called in many countries) is a way of getting more light out of the day by advancing clocks by some hour during the summer. During Daylight Saving Time, the sun appears to rise one hour later in the morning, when people are usually asleep anyway, and sets one hour later in the evening, seeming to stretch the day longer.

Step 1: Open Administration/Management webpage.

Step 2: Enable parameter **Daylight Saving Time** in Time/Date.

Step 3: Set **offset:** "-60" means advancing 60min, "60" means delaying 60min.

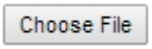
Step 4: Set starting Month/Week/Day/Hour in **Start Month/Start Day of Week Last in Month/Start Day of Week/Start Hour of Day**, analogously set stopping Month/Week/Day/Hour in **Stop Month/Stop Day of Week Last in Month/Stop Day of Week/Stop Hour of Day.**

Step 5: Click the **Save** button to keep changes and then click **Reboot** button for changes to appear.

8.11 Upgrade Firmware

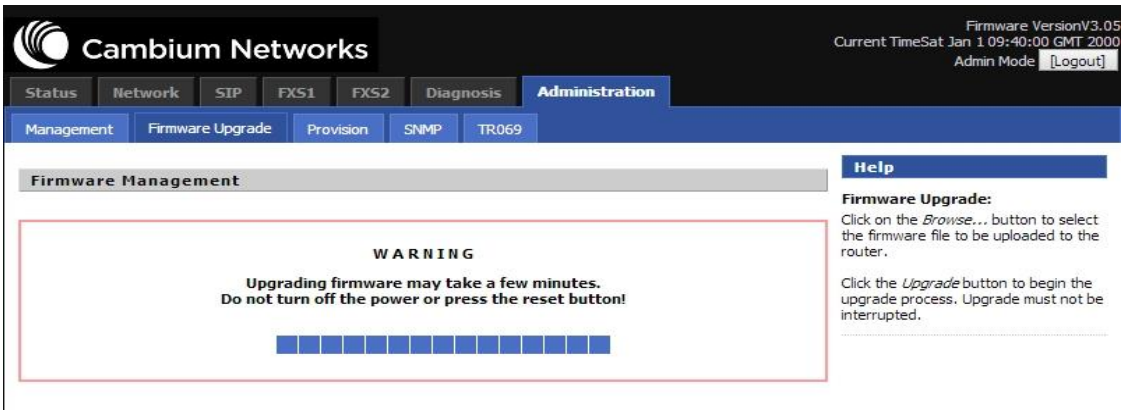
Function is to upgrade firmware in local.

Step 1: .Open Administration/Firmware Upgrade webpage

Step 2: Press  to browse a firmware file

Step 3: Press to start upgrading

Step 4: When upgrading, a warning prompt appears as shown below:



8.12 Password Control

Function is to reset password.
Step 1: Open **Administration=>Management** webpage
Step 2: Choose password type
Step 3: Input current password (Original Password, default is “admin” for admin level and user for user level),
Step 4: Input new password in New Password and Password Confirm.
Step 5: Click **Save** Settings button to and then Click **Reboot** button for the changes to appear.

8.13 Web Access

User can use the two parameters in Web Access to control WAN web login or login port.
WAN Interface Login is to disable/enable user access to web via WAN port;
Web Login Port is to set login port.

8.14 System log

User can view system log in local or in remote.
In local:
Step 1: Open **Administration=>Management** page, System Log Setting column.
Step 2: Choose log level from INFO and Debug, in INFO level, C3VOIP-150 records INFO log and in the Debug level, C3VOIP-150 records all debug information.
Step 3: Click the **Save** button to save and then click the **Reboot** button for the changes to appear.
In remote:
Step 1: Open **Administration=>Management** page, System Log Setting column.
Step 2: Fill system server IP Address or domain name into Syslog Server.
Step 3: Choose log level from INFO and Debug. In INFO level, C3VOIP-150 records INFO log and in Debug level C3VOIP-150 records all the debug information.
Step 4: Click the **Save** button to save and then click the **Reboot** button for the changes to appear.

9. Software Features

- Supports SIP V2.0（RFC 3261/RFC3262）
- Supports G.711 (A-Law, μ-Law), G.723, G.722 and G.729A/AB Codes
- Supports two RJ45 10/100M that one is RADIO (WAN) port and another is LAN port.
- Supports two RJ-11 for FXS port to connect your analog phone
- Supports IP address assignment using PPPoE, DHCP and Static IP
- Supports NAT traversal (Static NAT Route or by STUN)
- Supports Voice Activity Detection(VAD) ,Comfort Noise Generation(CNG) and Echo cancellation
- Supports jitter buffer for smooth voice reception
- Supports direct IP to IP dialing without registration
- Supports complementary features such as Call hold, Call waiting, Call forwarding, Call Transfer, Call Block, Hotline, Message Waiting Indicator and DTMF Ready (In-band, RFC2833 and SIP INFO) etc.
- Supports MAC address cloning
- Supports IEEE802.1Q VLAN/802.1P and IP TOS
- Provides easy auto and manual provisioning through manual operation (Web interface and IVR-driven interface) or auto provisioning via TFTP or HTTP
- Supports syslog client.

10. Third Party Software

The software may contain one or more items of Third Party Software supplied by other third - party suppliers. The terms of this Agreement govern your use of any Third-Party Software UNLESS A SEPARATE THIRD - PARTY SOFTWARE LICENSE IS INCLUDED, IN WHICH CASE YOUR USE OF THE THIRD - PARTY SOFTWARE WILL THEN BE GOVERNED BY THE SEPARATE THIRD - PARTY LICENSE.

10.1 OpenSSL

Copyright (c) 1998-2014 The OpenSSL Project. All rights reserved.
Redistribution and use in source and binary forms, with or without modification, are permitted provided that the following conditions are met:

1. Redistributions of source code must retain the above copyright notice, this list of conditions and the following disclaimer.
2. Redistributions in binary form must reproduce the above copyright notice, this list of conditions and the following disclaimer in the documentation and/or other materials provided with the distribution.
3. All advertising materials mentioning features or use of this software must display the following acknowledgment:
   "This product includes software developed by the OpenSSL Project for use in the OpenSSL Toolkit. (http://www.openssl.org/)"
4. The names "OpenSSL Toolkit" and "OpenSSL Project" must not be used to endorse or promote products derived from this software without prior written permission. For written permission, please contact openssl-core@openssl.org.
5. Products derived from this software may not be called "OpenSSL" nor may "OpenSSL" appear in their names without prior written permission of the OpenSSL Project.
6. Redistributions of any form whatsoever must retain the following acknowledgment:
   "This product includes software developed by the OpenSSL Project for use in the OpenSSL Toolkit (http://www.openssl.org/)"

THIS SOFTWARE IS PROVIDED BY THE OpenSSL PROJECT "AS IS" AND ANY EXPRESSED OR IMPLIED WARRANTIES, INCLUDING, BUT NOT LIMITED TO, THE IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE ARE DISCLAIMED. IN NO EVENT SHALL THE OpenSSL PROJECT OR ITS CONTRIBUTORS BE LIABLE FOR ANY DIRECT, INDIRECT, INCIDENTAL, SPECIAL, EXEMPLARY, OR CONSEQUENTIAL DAMAGES (INCLUDING, BUT NOT LIMITED TO, PROCUREMENT OF SUBSTITUTE GOODS OR SERVICES; LOSS OF USE, DATA, OR PROFITS; OR BUSINESS INTERRUPTION) HOWEVER CAUSED AND ON ANY THEORY OF LIABILITY, WHETHER IN CONTRACT, STRICT LIABILITY, OR TORT (INCLUDING NEGLIGENCE OR OTHERWISE) ARISING IN ANY WAY OUT OF THE USE OF THIS SOFTWARE, EVEN IF ADVISED OF THE POSSIBILITY OF SUCH DAMAGE.